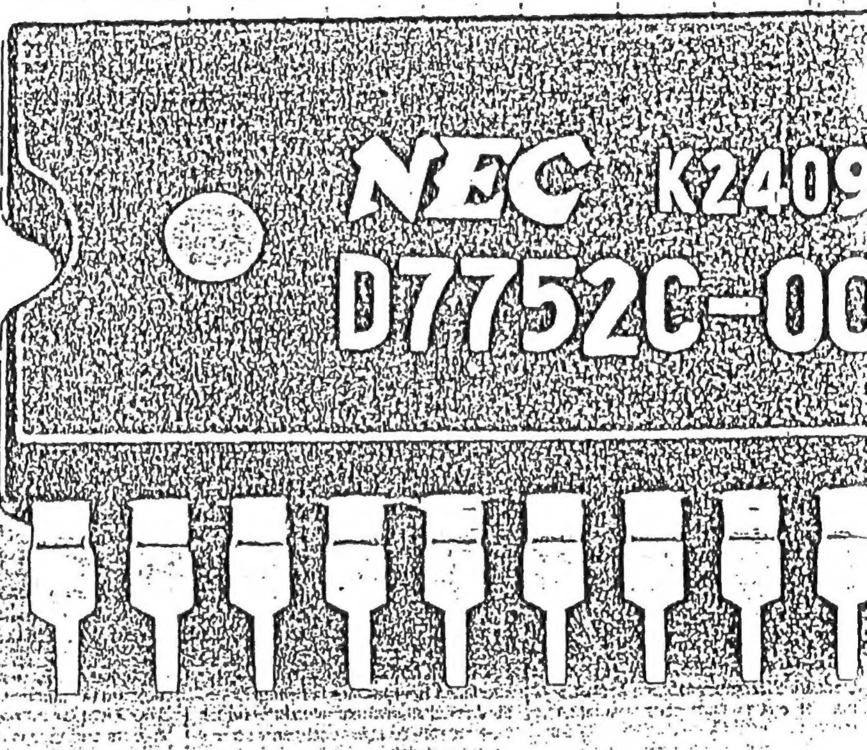


GROS PLAN SUR UN DIALOGUE HOMME-MACHINE

NEC

NEC ELECTRONICS FRANCE

Tour Chenonceaux
204, Rond Point
du Pont de Sèvres
92516 - Boulogne-Billancourt
Tél. 620.64.00

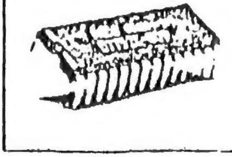
Un dialogue avec votre machine était jusqu'à ce jour du domaine de l'inaccessible. Maintenant grâce à NEC vous pouvez parler à votre machine et elle vous écoutera.

PARLER: 2 VLSI monochips de synthèse de la parole, à débit variable, ce qui permet différentes définitions de la voix reproduite.

Deux produits sont actuellement disponibles:
- le µPD 7751, NMOS, simple à mettre en œuvre, qui fonctionne en mode "ADPCM" entre 14 et 20 Kbits/sec., offre un temps de parole de 36 sec. maximum pour une capacité de mémoire externe adressable maximale de 512 Kbits.
- le µPD 7752, CMOS, qui fonctionne en mode "Formant" entre 1,2 et 5,6 Kbits/sec. maximum pour 63 mots de 512 mots avec l'appoint d'une mémoire externe.

ECOUTER: Les 3 VLSI disponibles, µPD 7760, 7761 et 7762 permettent de réaliser un ensemble fonctionnel de reconnaissance de la parole, dont le temps de réponse est de 0,7 sec. Les mots isolés, provenant d'un locuteur, sont reconnus dans plus de 90% des cas. 512 mots peuvent être entrés sur une mémoire de 64 K octets. Cet ensemble, connectable à n'importe quel système, offre les interfaces suivants: 8 bits parallèle, RS 232/V 24 et US série.

Grâce à notre technologie, dialoguez, d[...] aujourd'hui, avec votre machine.



SPEECH-SYNTHESIS μPD7751

SPEECH-SYNTHESIZING TECHNIQUE

ADPCM; ADAPTIVE DIFFERENTIAL PCM

FEATURES

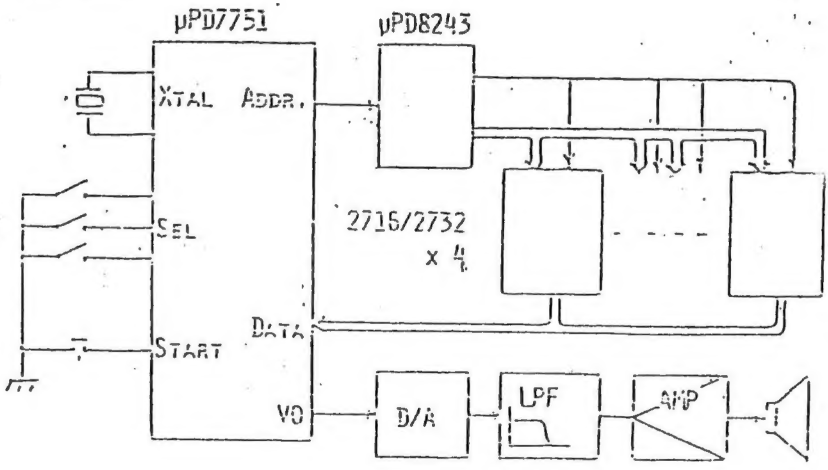
14K BPS - 20K BPS DATA RATE

HIGH-QUALITY SPEECH

8 MESSAGES

N-MOS, 40PIN DIP

SYSTEM DIAGRAM



μPD7751C

(LSI for ADPCM Voice Synthesis)

USER'S MANUAL

1. Features of μPD7751C

The μPD7751C is an LSI for voice synthesis, based on the waveform coding system. Although the μPD7751C uses the relatively high bit rate of 14 kbits/s to 20 kbits/s, natural voice quality can be obtained, and its development may be made in a relatively short period of time. Therefore, this chip is ideally suited for manufacture of small-quantity wide variety equipments, and for designing and in the process of voice applied equipming devilopment, and planning evaluation.

Features

- ADPCM Decoding Function (pseudo-ADPCM in detail)

- Operation on +5V Single Power Supply.

  (including peripheral circuits only with +5V single power source.)

- Eight Messages Selection

  Permits easy expansion to more than eight messages at the system level.

- Variable Bit Rate - 14 k bps to 20 k bps

  Bit rate corresponds to the sampling clock of 4 to 6 kHz (selection specified at time of analysis).

. Zero Signal Duration Compression Encoding Function

Efficiently encodes a message which includes a lot of no sound portion.

. Easy Voice Processing System

High fidelity voice --- Holds voice quality with high fidelity to actual voice.

Minimum talker dependence --- Minimized synthesis voice quality deviation by talker

Ease of analysis and data encoding --- Only a day of processing from recordign tape to ROM code. (10 sec speech)

Stable tone quality --- Elements and range of degradation, against raw voice are estimable..

. Background Music Mixing Capability

Capable of processing a plurality of tone sources with echo etc (but bit rate increases in this case).

2. μPD7751C Waveform Encoding System

Generally, the voice signal is transmitted as a complicated waveform containing the frequency components of 100 Hz to 10 kHz. Of these, the components below 2.5 to 3 kHz play an important role for transmission of the contets of ordinary messages. Here, an example of producing a voice synthesis using frequency components up to 2.5 kHz is given below.

First, the voice signal is sampled at twice the necessary frequency band (2.5 kHz) i.e. 5 kHz speed, according to the law called the "Nyquist low", thereby converting its amplitude value into digital data. For resolution of digital data, 8 bits are used in such a manner that noise does not present almost any problem as a voice signal.

At that stage, the voice waveform becomes 40-kbit/s (5 kHz with 8 bits/sample) digital data train.

From the nature of the voice signal, digitalized data is such that a difference in the values between sampling points adjacent to each other normally becomes a value smaller than its original value. By finding a difference between two adjacent data trains, therefore, conversion is made into a differential data train. This technique is known as the differential PCM (DPCM).

The majority of data of the voice waveform signal expressed in 8 bit form, as a result of differentiation, in essence becomes a small value expressed in less than 4 bits. So, it is possible to compress data per sample into 4 bits. In this

case, data volume is reduced by 20 kbits per second in 5 kHz x 4 bits. By reducing data of each sample to ... compression can be made to 20 kbits per second, but the original differential data train is for the most part can be expressed in 4 bits, but some data have 5 to 8 bits. So, when they are encoded into 4 bit code, the data of such large value cannot accurately be encoded.

If the original differential data has a value of more than 5 bits, the resolution for encoding (quantitized width) is made coarse, thereby encloding the data of large value within the range of 4 bits for the sake of solving this problem. This is called the adaptive differential PCM by the fact that the quantization width is varied according to the size of original data.

In the μPD7751, a technique is used by which the coarseness of quantization width is varied at ~~five steps in the ratio of two's exponent~~ (this is because the ratio of two's exponent can easily be calculated by the digital circuit). In so doing, proper quantization width (coarseness) is specified, to encode the entire range of original 8 bit differential data only in 4 bits. ... each quantitized width at 5 steps, and encoding range are shown in Fig.1 corresponding to the original 8 bit code,

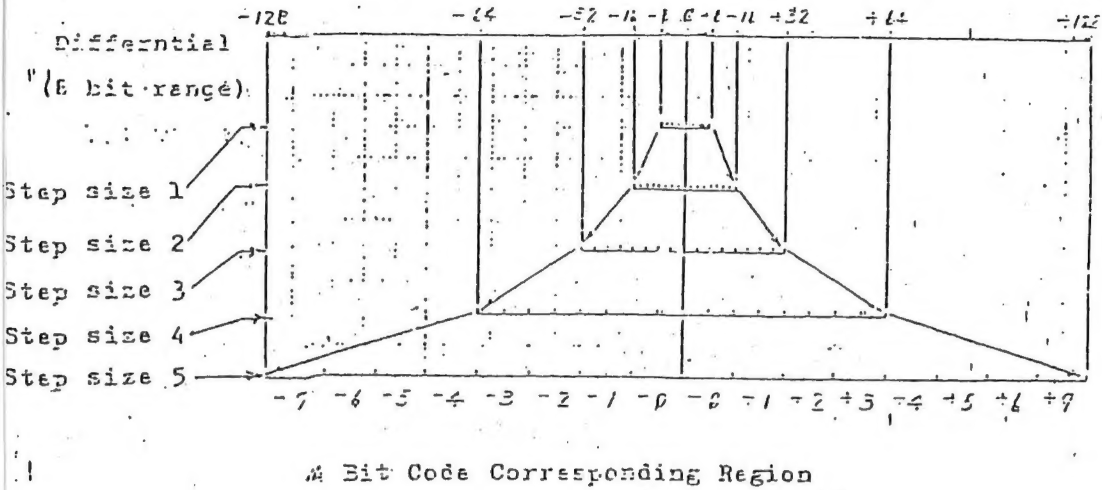


Fig. 1. Quantization Width (Step Size) and Corresponding Codes

In the μPD7755, the 8 bit DPCM voice data train is punctuated into groups called 'frame' for every 8 to 128 sampling points, and a proper step size is chosen from Fig. 1 so that the largest data can be encoded in 4 bits for each frame, and that the step size used at that time is encoded together with data for storing in the voice data ROM.

On the other hand, in the voice waveform, there is a period in a breathing spell or in a tone shrink point where the tone is rendered non-existent. This time will reach 10 to 20% during a message on an average. In this tone non-existent period, it is not necessary to encode or store the voice data. So, codes specifying these tone non-existent period are stored apart from the voice data.

As a result, the voice data ROM capacity is reduced to the extent of 10 to 20%.

The uPD7751 specifies quantization width based on the voice data compressed and encoded in this way, and decodes 8 bit differential data (DPCM) in consideration of the tone non-existent section specifying code.

Also, by making digital integration, original voice waveform data (8 bit PCM) is prepared, and then sent out (pins $VO_7 - VO_0$) in keeping pace with the sampling speed.

Note: For details of the encoding theorem, and the system configuration used in the uPD7751, refer to Appendix 1.

specifying input within the chip and then synthesezes and out messages selectively and separately. (For application to more than eight types of message, refer to Section 5.)

The uPD7751 is basically designed to be operated under control of other microprocessor. Because of the number of pins used and also because interface has to be made with various 4 bit or 8 bit microcomputers, the "Port type" input/output design instead of the "Bus type" design is employed. Terminals concertaining the interface with the input signal are:

- $\overline{SEL0}$ (35 pins), $\overline{SEL1}$ (36 pins), $\overline{SEL2}$ (37 pins): From eight types of message, one is selectively specified at active low input.
- $\overline{START}$ (6 pins): After $\overline{SEL}$ signal input, the low level pulse is input to start the syntheses output.
- $\overline{BUSY}$ (38 pins): This is the output for indicating the status of uPD7751 to the control. It becomes low level during message synthesising and output.

Timing relations between the input/output signals for these interface and the voice signal output are shown in Fig. 2.

The START signal is made low level simultaneously, or later than, the setting of the SEL signal. After maintaining the low level state for a period of more than 5 μS, ..., it has to be returned to high level . ... In the internal synthesizing operation of the actual uPD7751 starts from when the START signal has rised high from low level.

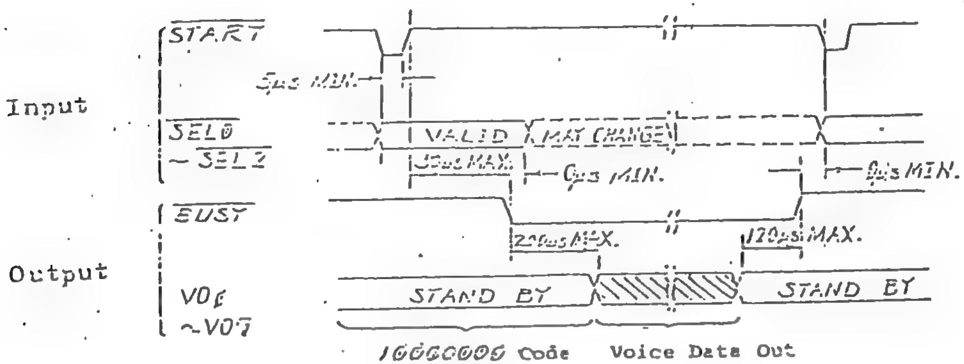


Fig. 2. Input Signal Timing

An example of the input circuit being used when operating the uPD7751 by mechanical key input is shown in Fig. 3.

Input pins SEL0 ~ SEL2 are pulled up to Vcc by 15 to 80kΩ ... . So, operation can be made by merely connect a switch to outside, as shown in Fig. 3.

The $\overline{BUSY}$ pin is used in open state, if it is not required.

In the circuit of Fig. 3, the $\overline{START}$ key is once turned ON. Then after synthesis operation starts when the key is turned OFF.

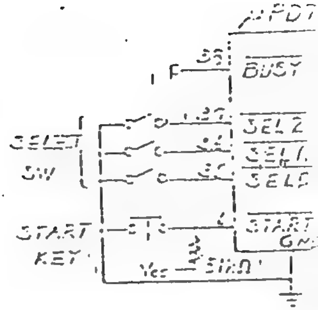


Fig. 3. Example of Key Input Circuit

Next, an example of interface with the I/O ports using 8255A in the 8 bit general-purpose micromputer is shown in Fig. 4.

In this example, 8255A is set in the mode 0, and is used by specifying the port A for the output, and the port B for the input. In the stand-by state, "H" has to be output to PA0 (ST... in advance.

To output voice, as shown in Fig. 4. set the message selection code to bits 3 - 1 and "L" to bit 0. ... the ... accumulator (Acc). Then, output to port A of 8255A by the OUT instruction. Next, increment the Acc (set bit 0 to "H") and again output it to 8255A. By so doing,

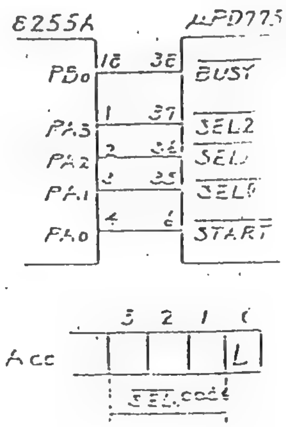


Fig. 4. Example of 825... Interface

the synthesizing operation is started from that time. Here, it should be noted that the signal to be sent to $\overline{SEL0 \sim 2}$ is active low.

4. Voice Data ROM Interface

The uPD7751 have to access to large-capacity ROM. So, because of the package pin count limitation the address driver (8243) is externally connected to obtain sixteen address outputs.

Also, address output can be made in two types of address modes so as to permit easy use of various ROMs and PROMs.

4.1 Chip Select Mode

This mode is used to externally connect less than four 16 Kbit or 32 Kbit PROM (2716/2732) as voice data ROM. An example of application circuit for connecting four 2732 is shown in Fig. 5.

Sixteen outputs (P40 - P73) of address driver (8243) are used for accessing PROM, but the internal address of 2732 are sent out to 12 outputs (P40 - P63) out of sixteen outputs On P70 - P73, the chip select signals of four 2732 chips are output. In this case, the directory field (refer to 4.3) is stored in the leading section (address 0000H) of ROM #0, and when the start signal ($\overline{START}$) is received by uPD7751, 0000H is first accessed thereby decoding the directory. (See 4.3 for information about the directory.)

4.2 Binary Address Mode

If the number of connecting chips of PROM are five or

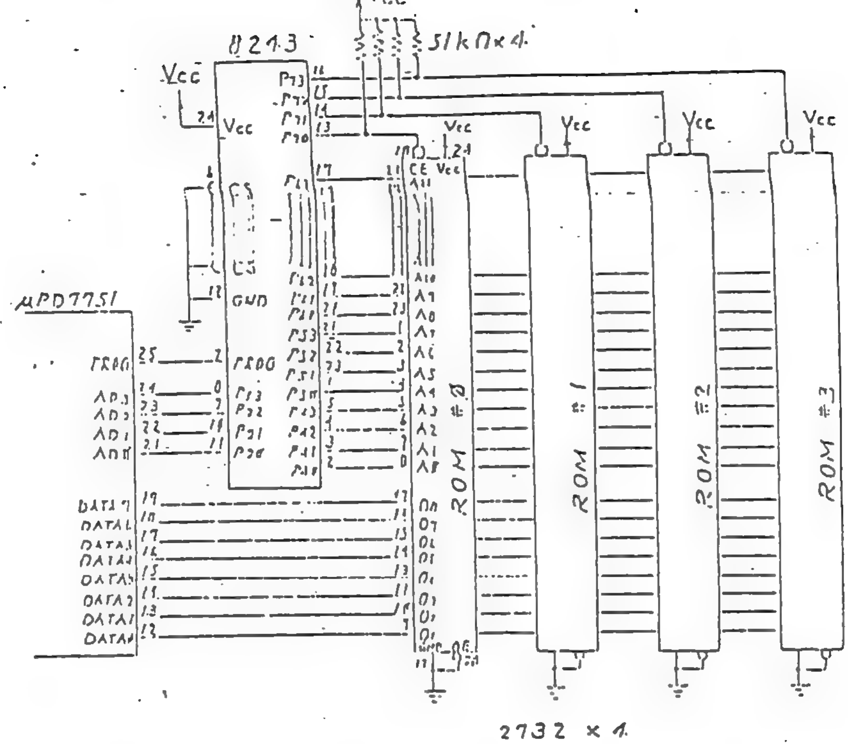


Fig. 5. Chip Select Mode ROM Interface

... when connecting ordinary mask ROM (2332, 2364, etc.), this mode should be selected. Any type of ROM with access time of less than 4 μS in the static operation can be applied.

An example of application using four 64 K ROM (2364) and to 256 K bit ROM addressing is shown in Fig.6. Binary addresses of 0 to 64 K can be output from the address driver (8243) on sixteen output lines, P40 - P73.

In this application, to obtain the chip select signal of four ROMs, high-order address outputs (3 outputs) of 8243 are decoded by the address decoder, 74LS138. When the 74LS138 (active low output) is used, programming is so made in the 2364 that CS pin is ... low active.

When selecting this mode, the ROM section in which the direc... is stored (the leading address section of ROM #0), must be located in E000H at the address output of 8243. So, care should be taken when making decoder output connection to the ROM chip. For this reason, Y7 output of 74LS138 is connected to ROM #0 chip, as shown in Fig. 6. The uPD77... is acccessed by sequentially incrementing the voice data RO... as from E000H so that ROM #0 becomes the address locations for E000H - FFFFH and ROM #1 for 0000H - 1FFFH. Chip sele... is made so that the last address becomes 5FFFH.

Also when using a large number of 2716 or 2732 PROMs, the design can be made as exemplified in Fig. 6. On the other

hand, when using a plurality of mask ROMs of 2316 or 2332, or when using only two 2364, the address decoder design can be simplified by utilizing the advantage that the CS pin of each mask is programmable.

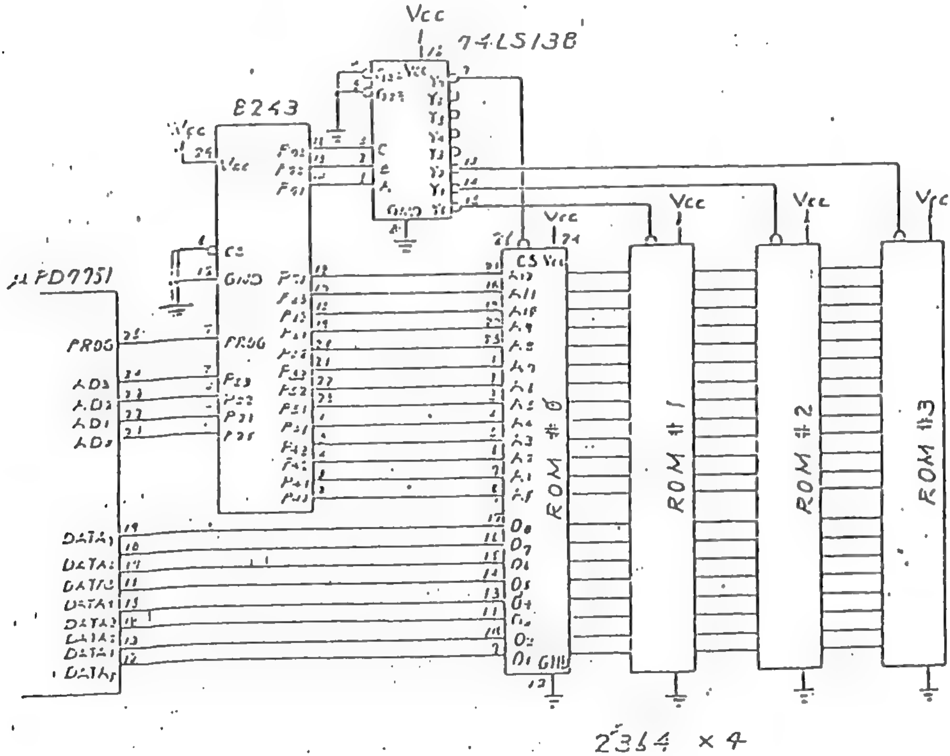


Fig 6. Binary Addressing Mode (256 K Bit ROM)

4.3 Directory

The uPD77P751 is capable of connection with a maximum of 512 Kbits as voice data ROM. But connection is presently limited to 256 Kbits (32 Kbytes) for reason of analyzing equipment capacity. Within this range of memory,

Capacity eight types of message may be stored, but to discriminate the data recorded area corresponding to each message, the selected message number, and the table corresponding to the message data recorded ROM address are written in the leading address section of the ROM.

In addition to this table, sampling frequency, address mode designation, etc. are recorded, and the 00H - 2FH section of the ROM #0 performs the directory area. The uPD7751 accesseo the directory section first when syntyesis starts.

As described in paras. 4.1 and 4.2, the directory area has to be placed in 0000H ~ in the chip select mode, and in E000H~ area in the binary address mode.

5. Method of Increasing the Number of Messages

The μPD7751 itself is able to select only eight types of massages since there are three input pins ($\overline{SEL0}$ - $\overline{SEL2}$) for message selection. EXpansion of message count can be easily accomplished with a Bank switching organization of voice data ROMs. When the Bank switching is applied, massage count can expand infinitely.

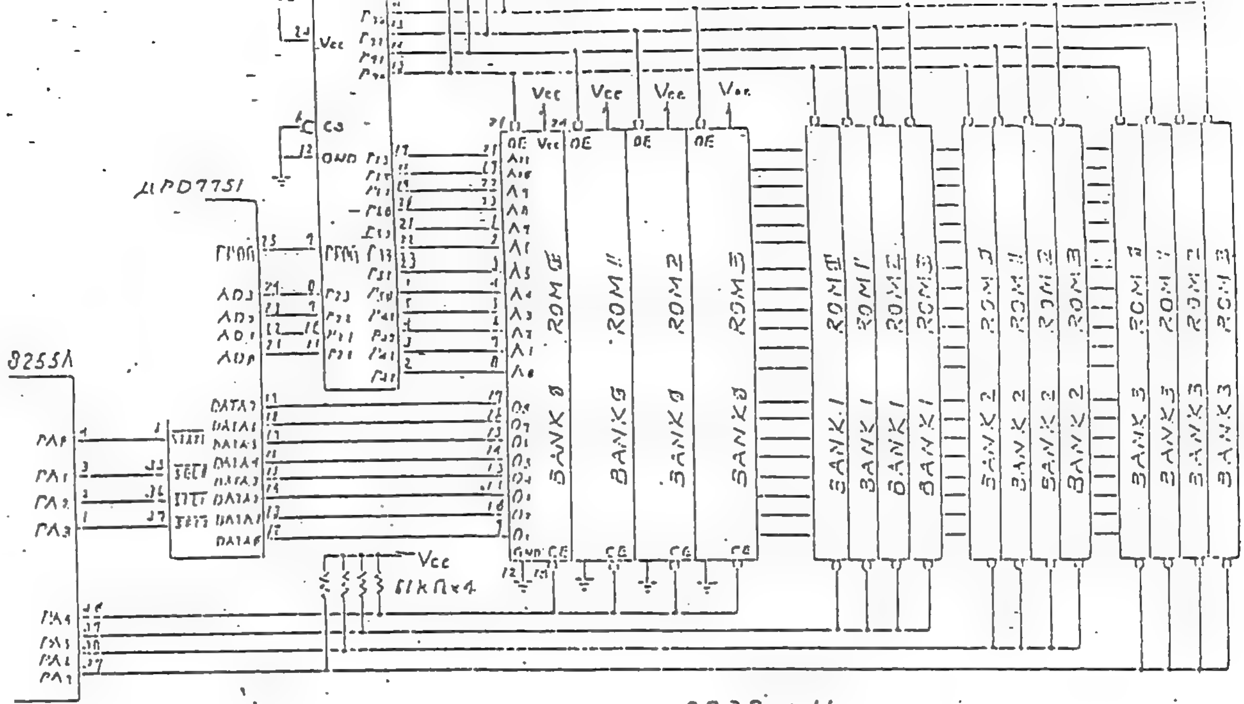
Fig. 7 shows a case where addressing is made in the chip select mode from the μPD7751 through 8243 and with four 32 K PROMs as one bank and four sets of bank (BANK0 - 3) are directly switched by 8255A port A. Each bank has 128 kbit capacity, and hence is able to register eight types of message. So, the total capacity is 512 kbits, 32 messages.

The capacity and the number of messages (less than eight types) in each bank can arbitrarily be set so that it is possible to increas or decrease the number of ohips of ROM per bank. For operation of the μPD7751C, each bank is totally independent from other banks other banks. Therefore the design such that the address mode, ROM chip used, sampling frequency, and other condition is varied for each bank, is available.

In Fig. 8, an application example of decoding the bank accessing signal (PA4 - PA7 of 8255A) by 74LS138 is shown. Each bank access eight 2732 (256 kbits) or less in the binary address mode, thereby selecting eight banks with the outputs of 8255A PA4 - PA6. Thus, the maximum number of messages are 64 types, and memory capacity can be expanded up to 2 M bit.

In the application of such bank switching system with 8255A, when bank switching signal varies during message synthesizing and

output process, necessary voice ROM data cannot be read. So, the 4 bit output of PA4 - PA7 of 8255A has to be stable from when the start signal is given till the busy signal is turned off. This point has to be carefully noted.



2732 x 16

Fig. 7. ROM Interface for 32 Message Selection

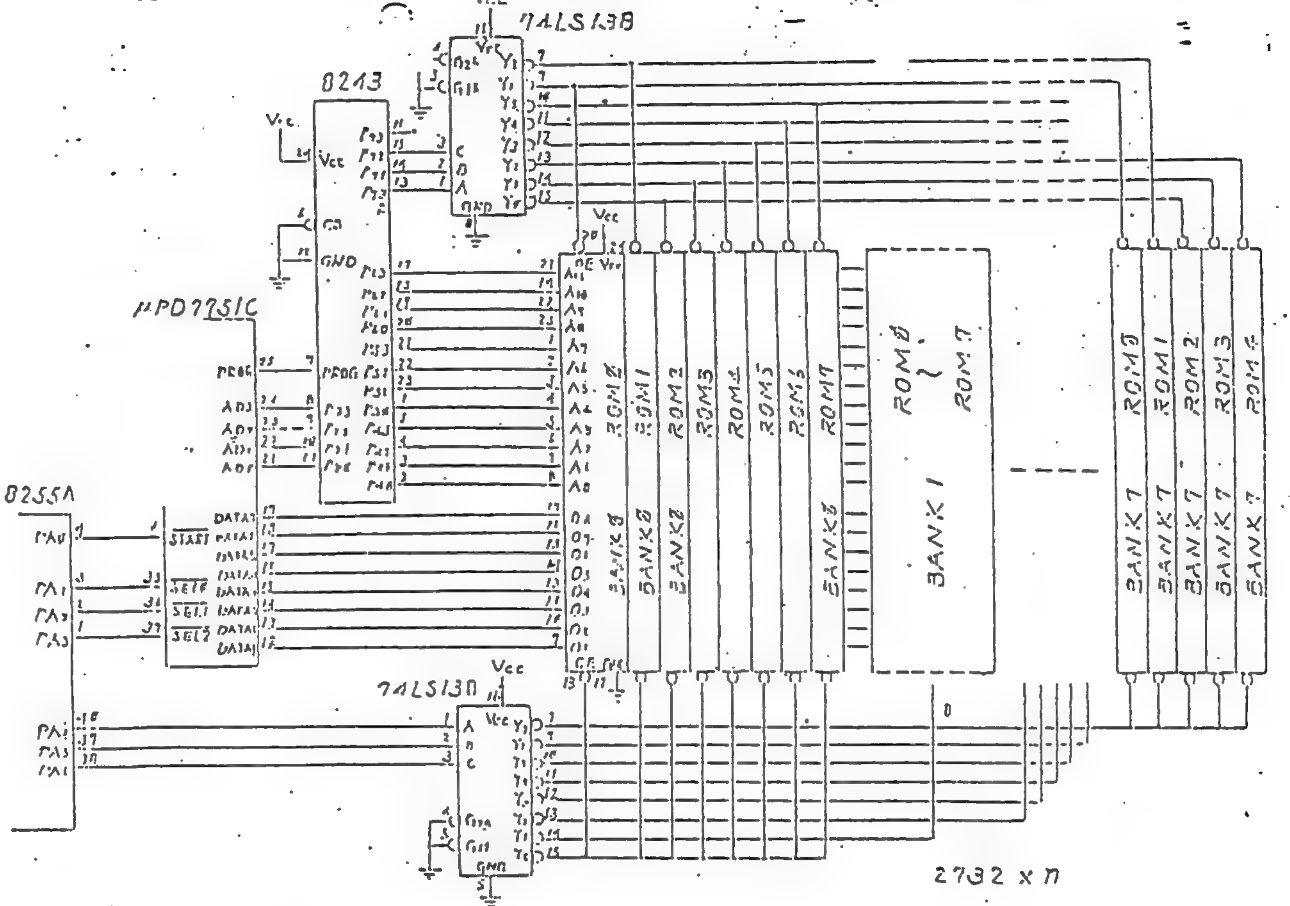


Fig. 8. Full Binary Decoding ROM Interface

6. Voice Output Interface

The μPD7751 outputs synthesis voice signal to pins VO_0 - VO_7 in the 8 bit offset? binary code, in synchronism with sampling frequency indicated in the ROM code.

After converting VO output into an analog signal by the D/A converter, it is necesoary the D/A back into signal passing via through a low-pass filter for cutting off the noise of the sampling frequency. Generally, a power amplifier is connected to the output ... of the filter, so as to obtain speaker output. The block diagram of output interface is shown in Fig. 9.

Fig. 9. Voice Output Interface

6.1 D/A Converter

The 8 bit D/A converter can be an IC type product (μPC624, etc.), which is easily obtainable at low cost. It is therefore very convenient to utilize such IC for this purpose.

Fig. 10. D/A Converter Circuit Using μPC624

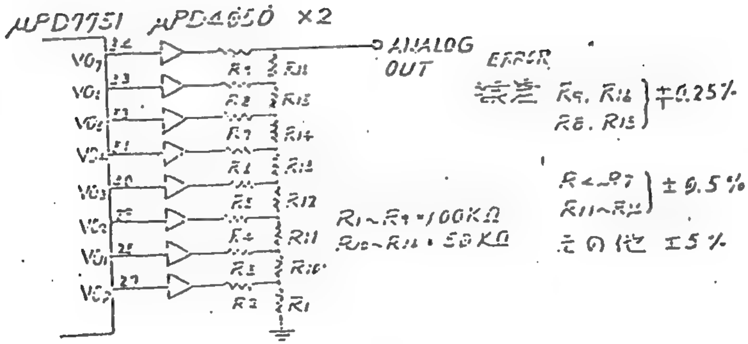


Fig. 11 D/A Converter with R-2R Ladder

An example of circuit using uPC624 is shown in Fig. 10. The analog signal which is output to I_{out} varies between 0 and +5 V. When no voice is issued, or when in the stand-by state, 10000000 (80H) is output into VO pin of μPD7751. So, the analog output is at about 2.5V level. At the time of voice synthesizing the voice waveform varies with the amplitude of ±2.5V from this level.

To use the μPC624, it is necessary to employ -5V power source, in addition to the +5V power source. When it is necessary to operate the voice synthesis system on a single source of +5V, the circuit using a resistor ladder (Fig. 11) is most proper. This circuit is a R-2R ladder type converter constructed with discreet resistors. CMOS buffer IC, 4050 is used for holding the inconsistency in the output resistance and output level of the μPD7751 to a minimum. Just as in the case of a circuit using the uPC624, the analog signal with amplitude of ±2.5 V can be obtained from the no voice output level of +2.5V being transmitted. But since the output impedance is relatively high (50 kΩ), it is important to pay special design attention to the rear-stage filter.

.2 Low-Pass Filter

The signal digitally analyzed and synthesized by the sampling

system can only reproduce the frequency range less than 1/2 the sampling frequency. The component higher than the 1/2 frequency acts as a deleterious noise. So, if D/A output is directly amplified, a disturbing voice with poor signal-to-noise ratio may result. Therefore, it is necessary to pass the output of D/A converter through a low-pass filter which elimunates the component higher than 1/2 the sampling frequency.

Steeper the characteristics of this filter, more highly desirable. The cutoff characteristics above 48 dB/oct. is ideal for the filter, but the design of this filter may become extremely complicated. So, a filter with 24 dB/oct characteristics, which stand the use in most cases is introduced below as a filter suited for the uPD7751.

Fig. 12 shows an active filter with Butterworth attenuation characteristics of 24 dB/oct, allowing for operation on +5V single power source.

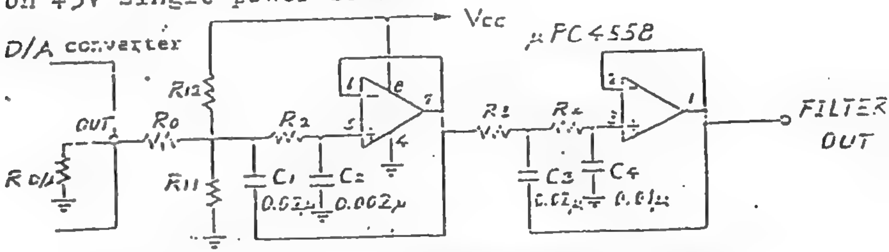


Fig. 12. 24dB/oct Low-Pass Filter Circuit

The supply voltage of this circuit is 5 V. So, input and output voltage amplitude is limited to ±0.5 V or so, when considered from the specification of μPC4558 operation. Therefore, it is probable in some cases that the output of the D/A converter mentioned in the preceding paragraph cannot be directly input.Note) Table 1 gives a tabulation of constants of each section of the filter in consideration of this point.

Table 1 indicates proper constants when the sampling frequency (f_{SMPL}) of 4, 5, and 6 kHz is used..

To obtain optimum characteristics, it is necessary to use accurate values of C and R, calculated in the table, but as those for uPD7751, the approximate values of C, R that can easily be obtainable which are indicated in parenthesis ( ) are practical.

When the filter circuit is used at the constants given in Table 1, the voice waveform signal of 2.5 V ±2.5 V, to be output from the D/A converter, is attenuated to the voice amplitude of ±0.33 V, with 2.5 V as mean level, thus meet the operating condition of the uPD4558. The characteristics of the filter are such as shown in Fig. 13. As the gain in the pass band range is 0 dB, the same voice signal (0.40 - 0.66 Vpp) as that at the input can be obtained at the output.

Table 1. 24dB/oct Butterworth Characteristics LPF Constant Table (Unit: kΩ)

D/A	f_{SMPL}	f_c	R_C	R_{11}, R_{12}	R_2	R_3	R_4
Fig.10	4kHz	1.8kHz	47.00 (47)	17.35 (18)	26.45 (27)	2.911 (3.0)	13.43 (13)
	5kHz	2.2kHz	47.00 (47)	13.75 (13)	21.64 (22)	2.382 (2.4)	10.92 (11)
	6kHz	2.7kHz	47.00 (47)	10.93 (11)	17.63 (18)	1.941 (2.0)	8.950 (9.1)
Fig 11	4kHz	1.8kHz	0	17.35 (18)	26.45 (27)	2.911 (3.0)	13.43 (13)
	5kHz	2.2kHz	0	13.75 (13)	21.64 (22)	2.382 (2.4)	10.95 (11)
	6kHz	2.7kHz	0	10.93 (11)	17.63 (18)	1.941 (2.0)	8.950 (9.1)

The figure in parenthesis denotes an approximate value by commercially available E24 type resistor.

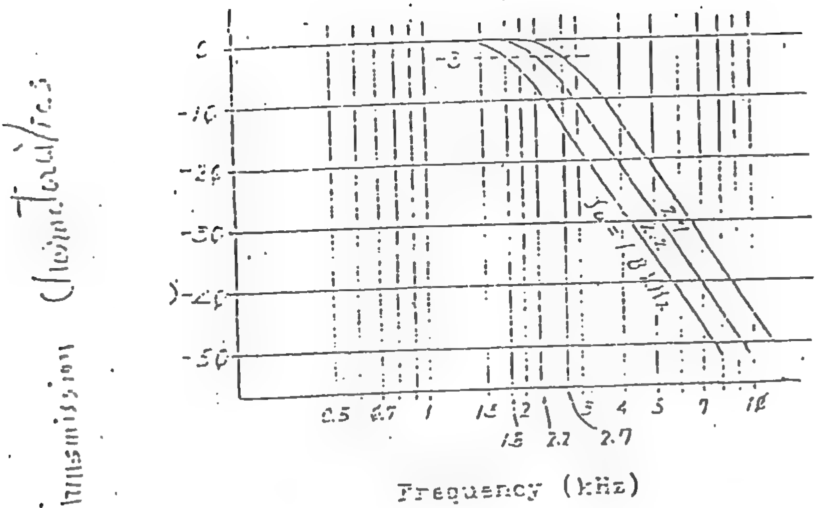


Fig. 13: Low-pass Filter Characteristic Curves

6.3 Power Amplifier

The voice signal obtained as the output of the filter described in the preceding paragraph is exactly the same as the ordinary analog waveform. So, it can be connected to almost any power amplifier when required. Here, an example of low voltage operation power amplifier operating on +5 V single power source is shown in Fig. 14.

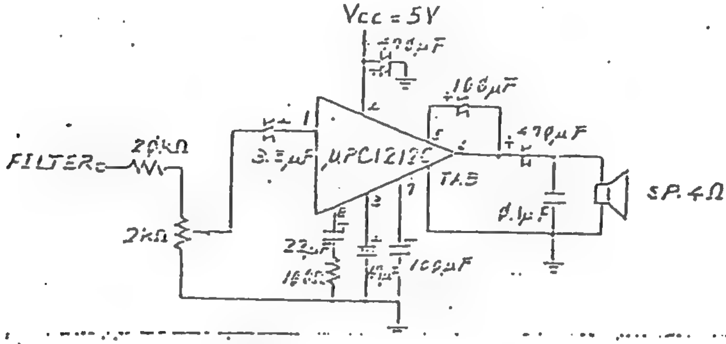


Fig. 14. 0.7W 5V Power Amplifier

Fig. 14. 0.7W 5V Power Amplifier

The uPC1212C is an IC operating in the supply voltage range from 4.5 V to 7.0 V. When the supply voltage is 5.0 V, it is possible to obtain the output of 0.7 W (T.H.D = 10%) using a 4-ohm speaker.

The uPC1212C has the voltage gain of more than 40 dB. When the output of the low-pass filter shown in Fig. 12 is directly connected, the input becomes excessive. So, a resistor for attenuation is inserted in the input section.

To operate the circuit of Fig. 14 on +5V power (Vcc) of the same digital system as the uPD7751, it is necessary to ensure that the noise generated from the digital system does not appear. When wiring the power amplifier, one-point grounding should be made of the input section, power supply bypass, and output section separately, as shown in Fig. 14.

7. Example of μPD7751 Voice synthesis System Design

An example of the entire circuit for the voice synthesis system using the μPD7751 is shown in Fig. 15. In this example, four chips of μPD2732 (128 Kbits) are connected as voice data ROM. The total output time of voice varies depending upon the sampling frequency involved, but it is approximately nine second when the sampling frequency is 4 kHz.

The circuit of each section is almost the same as described in preceding sections other pins of μPD7751 not described are as follows.
The operation clock of the μPD7751 is oscillated by connecting a 6.0MHz crystal to pins 2, 3. In undertaking the circuit design, pins 2, 3 of the μPD7751 and crystal should be connected to as close to each other as possible.

The RESET terminal (4 pin ...) externally has a 1μF capacitor provided to allow automatically initialize the inside of the μPD7751 when the power is turned ON. Also, Voice output can be stopped with this terminal externally forced low. So, it is possible to utilize this when cutting off the voice in the process of voice synthesis and outputing.

The pins (pins 1, 7, and 39) for internal testing should be unfailingly connected to the ground.

In the circuit of Fig. 15, the μPD4069 is inserted as a buffer of the D/A converter. Also, the constants of the low pass filter are set to the appropriate value when the sampling frequency (f_{SMPL}) is 4 kHz.

power in all cases so that the power for the digital circuit can be used. Also, by using PROM, write operation may be done quite easily. It is therefore ideal for evaluation of short delivery time small-lot equipments and also for evaluation of voice application system development.

(1) Ext. reset input

(2) START key

(3) Select sw

(4) Busy output

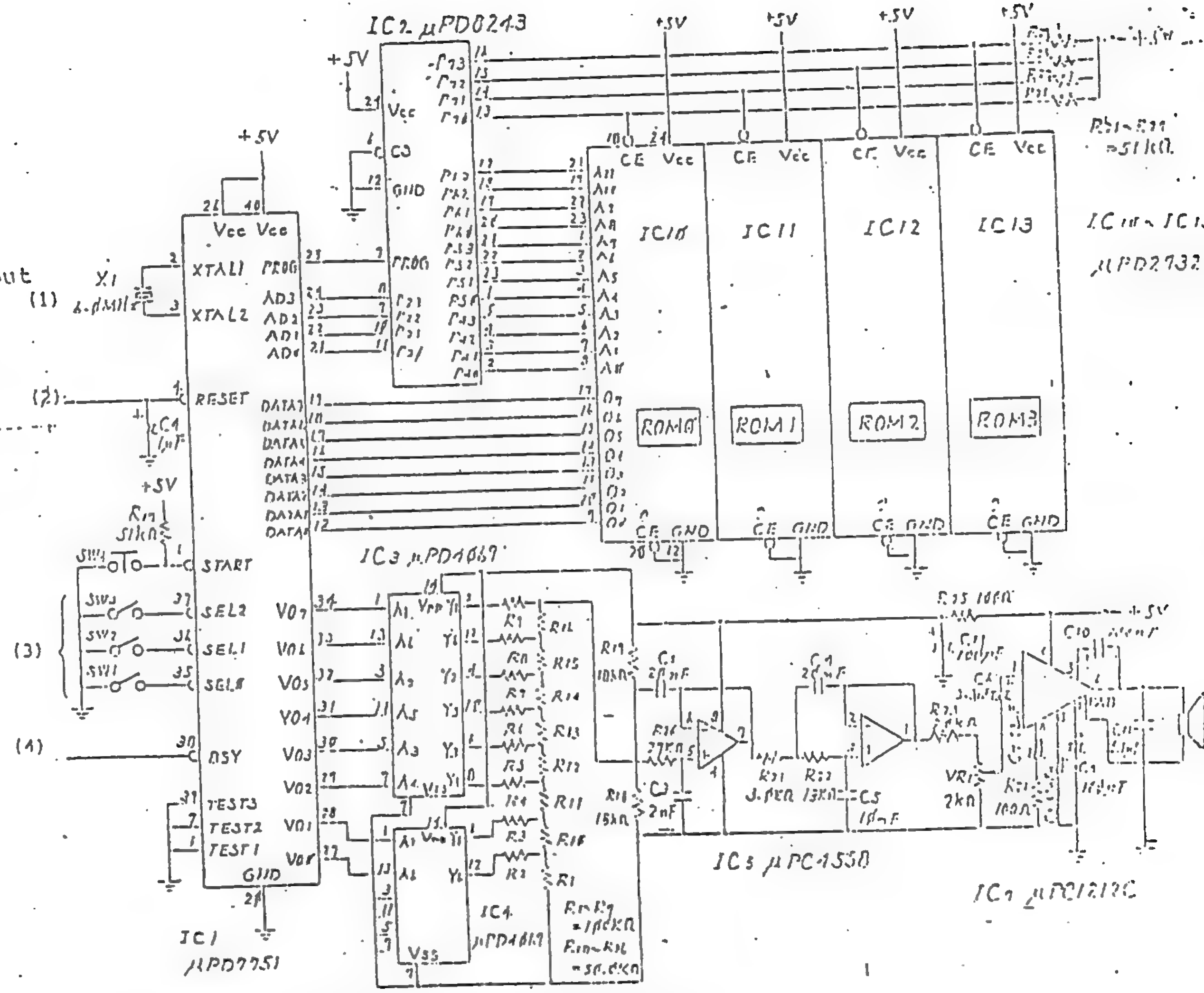


Fig. 15 Example of μPD7751C System

8. Development of Voice Code

As described in Section 1, the uPD7751C has the following features:

(1) A little change in hue of synthesis voice from original voice ... which makes it possible to keep the features of speech manner in raw voice.

(2) Capability to make analysis and encoding in a short period of time because encoding analysis and conversion can be made in a relatively simple manner.

The procedure of ROM code development for uPD7751C is as shown in the flow chart of Fig. 16. Major points are explained below along with this flow chart.

8.1 Message Preparation

The synthesis voice has insufficient reproduction frequency band as compared with natural voice. Efforts should be made to avoid use of confusing messages. If messages are so prepared that common phrases can be used, ROM saving can be realized to a significant extent.

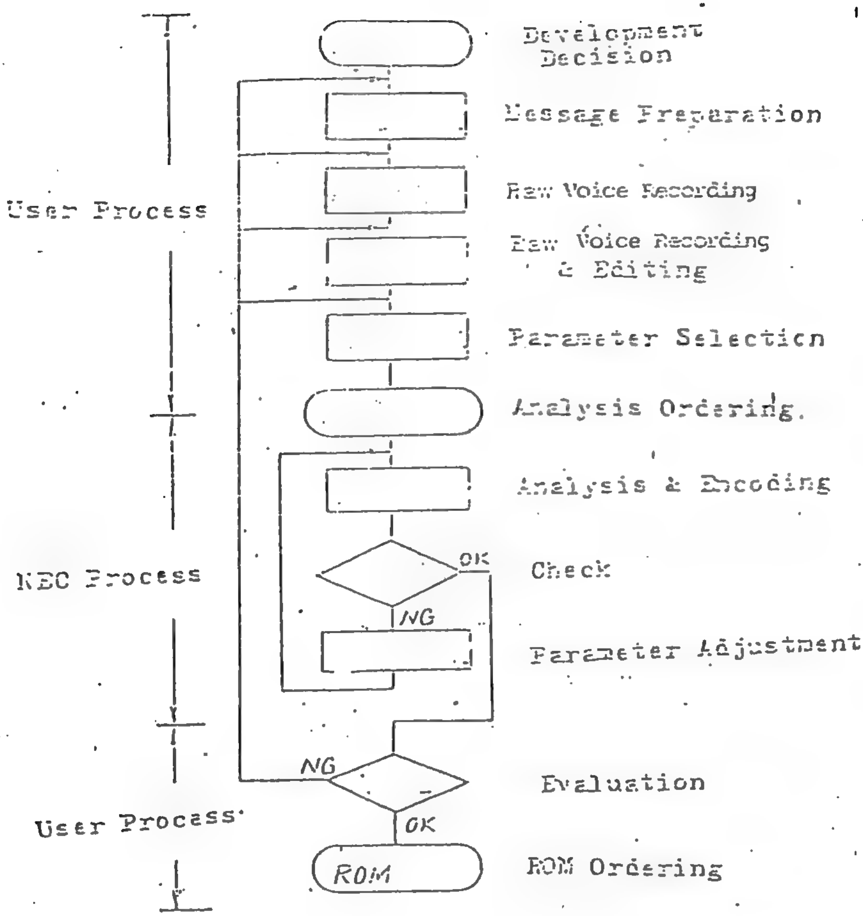


Fig. 16. ROM Code Development Chart

8.2 Raw Voice Recording

As the synthesis voice has relatively narrow effective dynamic range, better voice quality can be expected by speaking in a flat intonation.

In the synthesis voice, it is sometimes probable that noise in the row voice is emphasized. Care should therefore be taken to ensure that noise (hum noise in particular) in the message speaking is minimized. For this reason, it is advisable to record the row voice on the open reel tape using a formal recording studio. Also, it is suggested to have the same message recorded several times to cope with unexpected noise mixture and to make voice quality selection.

8.3 Raw Voice Evaluation and Editing

Juding noise, flat speaking, and voice quality from the recorded tape, the message finally selected is edited on the cassette tape As a cassette tape, a chrome tape should be used from the signal-to-noise ratio point of view, and recording should be made at a level somewhat high so that the peak level is at +5 to +8 dB. For easier analysis, it is necessary to insert a proper non-tone section sequentially, as shown in Fig. 17.

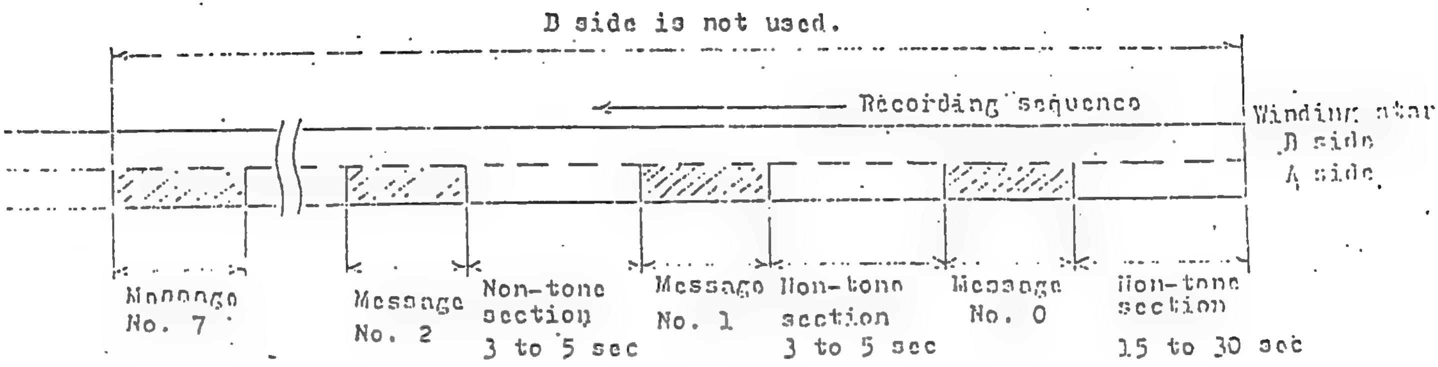


Fig. 17. Method of Editing Raw Voice Cassette Tape for Analysis

If there are a plurality of banks (more than 8 messages), a pause of 15 to 30 seconds should be provided once again after a BANK recording (Fig. 17). The next bank message should then be recorded.

8.4 Parameter Selection

Prior to ordering the analysis, it is necessary to decide on parameters for analysis. Parameters necessary for analysis are the address output mode, sampling

frequency, pre-emphasis, and data ROM file name.

(1) Address Output Mode and Number of ROM Used

Either the chip select mode or binary address mode is selected, and using bit capacity in a BANK is specified.

(2) Sampling Frequency

Either 4, 5, or 6 kHz has to be specified.

At 6 kHz, the combined tone will be nearly the frequency range of telephone, but at 4 kHz, the synthesized freque range is below 2 kHz. In that case, clarity is somewhat reduced: The selection of frequency cannot be made definitely, but the rough yardstick for it is as given in Table 2.

Sampling Frequency	Syntheses Freq Range	Applied Message	Voice Quality	Bit Rate (bps)
4 kHz	~ 1.8kHz	Male Sentence Black	Heavy and Confined	14 k 16 k
5 kHz	~ 2.2kHz	Male Word Block Female Sentence Block	Slightly heavy and confined	17 k 20 k
6 kHz	~ 2.7kHz	Male, Female Word Block	Telephone level or so	20 k 24 k

Table 2. Sampling Frequency and Application Messages

Care should be taken to the fact that the bit rate becom somewhat higher if the original tape has considerable noi

(3) Pre-emphasis

It is sometimes probable that better voice quality can be expected by previously adjusting the frequency response (pre-emphasis) prior to analysis, depending upon the sound environment such as reflection of place at which voice synthesis device is ultimately used, and the characteristics of the speaker and speaker box. When this is specified, the frequency component higher than 700 Hz can be emphasized, and the characteristics of +10 dB can be obtained at 2 kHz.

(4) Data ROM File Name

The analyzed data is registered in the mini Floppy disk (for PDA800) in the same format as the ROM code actually used. The file name (less than 7 characters) to be used at for this file must be specified.

(5) Analysis Ordering

After the decision of the above specifications all materials which is sent to NEC for analysis ordering are:

Raw voice cassette tape: Chrome tape, A side recorded, giving your company name, date, and ROM file name on the cassette half.

Message list : Itemized message listing together with message selection code in the sequence of recording.

Parameter designation : Using address mode, desired sampling frequency, and with or without pre-emphasis.

Be certain to clearly indicate your company name, ordering department, and the name of man in charge.

(6) Evaluation:

The result of analysis and encoding will be so indicated to you by recording the synthesis voice on a cassett half. The cassette will then be returned to you for evaluation. Also, the ROM code will be recorded in the specified file a mini Floppy disk (for PDA800) with the specified file name, and will be returned to you.

If no problem is encountered as a result of evaluation, you are requested to order PROM or mask ROM, as required.

If any problem arisses, you are requested to get in touch with our office, with your claims.